AMENDMENTS TO THE SPECIFICATION:

Please amend the paragraph starting at page 7, line 5 as follows:

In this example, the phase of analysis of the audio signal (block A1), shown in figure 2, comprises the following stages:

- shaping of the input signal (block 1),
- calculation of the temporal envelope (block 2),
- detection of temporal interpolation (block 3),
- detection of the audible signal (block 4),
- calculation of the temporal interpolation (block 5),
- calculation of the dynamic range of the signal (block 6),
- detection of an inaudible frame after a frame of higher energy (block 7),
- pulse processing,
- repetition of the pulse (block 9),
- calculation of the Fast ~~Fourrier~~ Fourier Transformation (FFT) on repeated pulse (block 10),
- calculation of the parameters of the signal used for the preprocessing before the FFT (block 11),
- preprocessing of the temporal signal (block 12),
- calculation of the FFT on processed signal (block 13),
- calculation of the signal-to-noise ratio (block 14),
- test of the Doppler variation of the pitch (block 15),
- calculation of the FFT on unprocessed signal (block 16),
- calculation of the signal-to-noise ratio (block 17),
- comparison of the signal-to-noise ratios with and without preprocessing (block 18),
- restitution of the result of the FFT with preprocessing (block 19),

- calculation of the frequencies and moduli (amplitudes of the frequential components (block 20),
- decision of the type of signal (bloc21),
- test of the 50 or 60 Hz (block 22),
- calculation of the dynamic range of the moduli in the frequential domain (block 23),
- suppression of the interpolation on the frequential data (block 24),
- suppression of the inaudible signal (block 25),
- calculation and validation of the pitch (block 26),
- decision if noise filtering or special effects, or continuation of the analysis (block 27),
- optional attenuation of the ambient noise (block 28),
- end of processing of the frame (block 29).

Please amend the paragraph starting at page 8, line 11 as follows:

The use of the Fast ~~Fourrier~~ Fourier Transformation (FFT) for the voice cannot be considered given the variability of the frequential signal; in fact the variation of the frequencies creates a spreading of the result of ~~said~~ the Fast ~~Fourrier~~ Fourier Transformation (FFT); the elimination of this spreading is made possible by means of the calculation of the variation of the pitch and by the application of the inverse variation of ~~said~~ the pitch on the temporal signal.

Please amend the paragraph starting at page 8, line 18 as follows:

Thus, the analysis of the vocal signal is carried out essentially in four stages:

- calculation of the envelope of the signal (block 2),

- calculation of the pitch and of its variation (block 12),
- application of the inverse variation of the pitch to the temporal signal (block 12),
- Fast ~~Fourrier~~ Fourier Transformation (FFT) on the preprocessed signal (block 13),
- optional elimination of the ambient noise before coding (blocks 23 to 28).

Please amend the paragraph starting at page 8, line 29 as follows:

Furthermore, a fifth threshold (block 15) makes it possible to carry out the Fast ~~Fourrier~~ Fourier Transformation (FFT) on the unprocessed signal as a function of the characteristics of the pitch and of its variation.

Please amend the paragraph starting at page 9, line 1 as follows:

A sixth threshold (block 18) makes it possible to retrieve the result of the Fast ~~Fourrier~~ Fourier Transformation (FFT) with preprocessing as a function of the signal-to-noise ratio.

Please amend the paragraph starting at page 11, line 1 as follows:

In the presence of a pulse, the repetition of the pulse (block 9) is carried out by creating an artificial pitch, equal to the duration of the pulse, in order to avoid the masking of the useful frequencies during the Fast ~~Fourrier~~ Fourier Transformation (FFT).

Please amend the paragraph starting at page 11, line 4 as follows:

The Fast ~~Fourrier~~ Fourier Transformation (FFT) (block 10) is then carried out on the repeated pulse by retaining only the absolute value of the complex number and not the phase; the calculation of the frequencies and of the moduli of the frequential data (block 20) is then carried out.

Please amend the paragraph starting at page 12, line 7 as follows:

The over-sampling, with a ratio of two, of the analysis frame is carried out by multiplying the result of the Fast ~~Fourrier~~ Fourier Transformation (FFT) of the analysis frame by the factor exp(-j*2*PI*k/(2*L_frame), in such a way as to add a delay of half of a sample to the temporal signal used for the calculation of the Fast ~~Fourrier~~ Fourier Transformation; the reverse Fast ~~Fourrier~~ Fourier Transformation is then carried out in order to obtain the temporal signal shifted by half a sample.

Please amend the paragraph starting at page 12, line 16 as follows:

After elimination of the variation of the pitch, ~~said~~ the pitch seems identical over the whole of the analysis window, which will give a result of the Fast ~~Fourrier~~ Fourier Transformation (FFT) without spread of frequencies; the Fast ~~Fourrier~~ Fourier Transformation (FFT) can then be carried out in block 13 in order to know the frequential domain of the analysis frame; the method used makes it possible to calculate rapidly the modulus of the complex number to the detriment of the phase of the signal.

Please amend the paragraph starting at page 12, line 23 as follows:

The calculation of the signal-to-noise ratio is carried out on the absolute value of the result of the Fast ~~Fourrier~~ Fourier Transformation (FFT); ~~said~~ the ratio is in fact the ratio of the difference between the energy of the signal and of the noise to the sum of the energy of the signal and of the noise; the numerator of ~~said~~ the ratio corresponds to the logarithm of the difference between two energy peaks, respectively of the signal and of the noise, the energy peak being that which is either higher than the four adjacent samples corresponding with the harmonic signal, or lower than the four adjacent samples corresponding with the noise; the denominator is the sum of the logarithms of all the peaks of the signal and of the noise; moreover, the calculation of the signal-to-noise ratio is carried out in sub-bands, the highest sub-bands, in terms of level, are averaged and give the sought ratio.

Please amend the paragraph starting at page 13, line 7 as follows:

This distinction is then made in block 15; in fact, tests are carried out on the Doppler variation of the pitch and on the frequency of the pitch; if the variation of the pitch is low or its frequency high, the processing is immediately followed by the calculation of the frequencies and of the moduli of the frequential data of the Fast ~~Fourrier~~ Fourier Transformation (FFT) (block 20); in the opposite case, the Fast ~~Fourrier~~ Fourier Transformation(FFT) is carried out without preprocessing (block 16).

Please amend the paragraph starting at page 13, line 14 as follows:

The calculation of the signal-to-noise ratio is then carried out in block 17, in order to transmit to block 20 the results of the Fast ~~Fourrier~~ Fourier Transformation (FFT) without preprocessing, the case of a zero variation of the pitch, or, in the opposite case to retrieve the results of the Fast ~~Fourrier~~ Fourier Transformation (FFT) with preprocessing (block 19).

Please amend the paragraph starting at page 13, line 20 as follows:

This distinction is made in block 18, in the following way:

- if the signal-to-noise ratio without preprocessing is higher than the signal-to-noise ratio with preprocessing, the results of the Fast ~~Fourrier~~ Fourier Transformation (FFT) are transferred to block 20,
- if the signal-to-noise ratio without preprocessing is lower than the signal-to-noise ratio with processing, the retrieval of the results of the Fast ~~Fourrier~~ Fourier Transformation (FFT) with preprocessing being carried out in block 19, the results obtained with preprocessing are then transferred to block 20.

Please amend the paragraph starting at page 13, line 32 as follows:

The calculation of the frequencies and of the moduli of the frequential data of the Fast ~~Fourrier~~ Fourier Transformation (FFT) is carried out in block 20.

Please amend the paragraph starting at page 14, line 1 as follows:

The Fast ~~Fourrier~~ Fourier Transformation (FFT), previously mentioned with reference to blocks 10, 13, 16, is carried out, by way of example, on 256 samples in the case of a shifted frame or of a pulse, or on double the amount of samples in the case of a centred frame without a pulse.

Please amend the paragraph starting at page 14, line 5 as follows:

A weighting of the samples situated at the extremities of the samplings, called HAMMING weighting, is carried out in the case of the Fast ~~Fourrier~~ Fourier Transformation (FFT) on *n* samples; on *2n* samples, the HAMMING weighting window is used multiplied by the square root of the HAMMING window.

Please amend the paragraph starting at page 14, line 9 as follows:

From absolute values of the complex data of the Fast ~~Fourrier~~ Fourier Transformation (FFT), there is calculated the ratio between two adjacent maximal values, each one representing the product of the amplitude of the frequential component and a cardinal sine; by successive approximations, this ratio between the maximal values is compared with the values contained in tables, containing this same ratio, for N frequencies (for example 32 or 64) distributed uniformly over a half sample of the Fast ~~Fourrier~~ Fourier Transformation (FFT). The index of ~~said~~ the table which defines the ratio closest to that to be compared gives, on the one hand, the modulus and, on the other hand, the frequency for each maximum of the absolute value of the Fast ~~Fourrier~~ Fourier Transformation (FFT).

Please amend the paragraph starting at page 14, line 20 as follows:

Moreover, the calculation of the frequencies and of the moduli of the frequential data of the Fast ~~Fourrier~~ Fourier Transformation (FFT), carried out in block 20, also makes it possible to detect a DTMF (Dual Tone Multi-Frequency) signal in telephony.

Please amend the paragraph starting at page 23, line 1 as follows:

Another method of synthesis consists in carrying out the reverse analysis by recreating the frequential domain from the cardinal sine produced with the modulus, the frequency and the phase, and then by carrying out a reverse Fast ~~Fourrier~~ Fourier Transformation (FFT), followed by the product of the inverse of the HAMMING window in order to obtain the temporal domain of the signal.

Please amend the paragraph starting at page 36, line 30 as follows:

More precisely, the device can comprise:

- analysis means making it possible to determine parameters representative of ~~said~~ the sound signal, ~~said~~ the analysis means comprising:
  - means of calculation of the envelope of the signal,
  - means of calculation of the pitch and of its variation,
  - means of application of the inverse variation of the pitch to the temporal signal,
  - means for the Fast ~~Fourrier~~ Fourier Transformation (FFT) of the preprocessed signal,

- means of extraction of the frequential components and their amplitudes from ~~said~~ <u>the</u> signal, from the result of the Fast ~~Fourrier~~ <u>Fourier</u> Transformation,
- means of optional elimination of the ambient noise by selective filtering before coding,